(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: **0 522 213 A1**

(12) **EUROPEAN PATENT APPLICATION**

(21) Application number: **91306317.8**

(22) Date of filing: **11.07.91**

(51) Int. Cl.[5]: **G10L 3/00, G10L 3/02**

(43) Date of publication of application: **13.01.93 Bulletin 93/02**

(84) Designated Contracting States: **BE CH DE DK ES FR IT LI NL SE**

(71) Applicant: **NATIONAL RESEARCH COUNCIL OF CANADA**
Montreal Road
Ottawa Ontario K1A OR6(CA)

(72) Inventor: **Hunt, Melvyn**
20 Briar Wood Close
Fareham,Hampshire, PO16 0PS(GB)

(74) Representative: **Cockayne, Gillian et al**
GEC Patent Department, GEC Marconi Limited, West Hanningfield Road
Great Baddow, Chelmsford Essex CM2 8HN(GB)

(54) **System for separating speech from background noise.**

(57) A digital signal processing system applies an adaptive filtering technique to sequences of energy estimates in each of two signal channels, one channel containing speech and environmental noise and the other channel containing primarily the same environmental noise. From the channel containing primarily environmental noise, a prediction is made of the energy of that noise in the channel containing both the speech and that noise, so that the noise can be extracted from the mixture of speech and noise. The result is that the speech will be more easily recognizable by either human listeners or speech recognition systems.

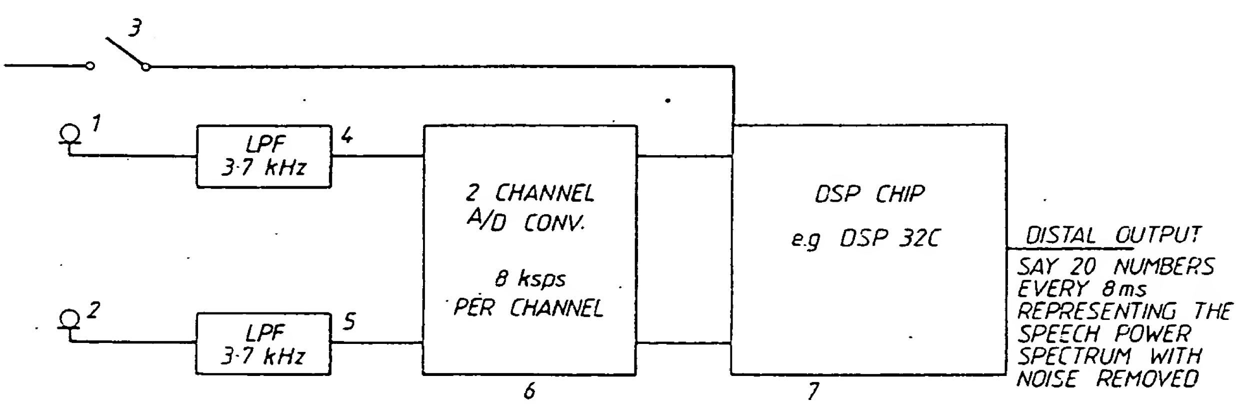


Fig.1.

EP 0 522 213 A1

Background of the Invention

1. Field of the Invention

The invention relates to a method of processing speech mixed with noise that are concurrently detected by a microphone in a noisy environment. In many situations where communication with machines by voice using automatic speech recognition would be desirable, the application of speech recognition technology is unsuccessful because the background noise interferes with the operation of the speech recognition system. Examples of such situations are helicopters, airplanes, battle tanks, automobiles, factories, postal centres and baggage handling centres. This invention also has potential application to a class of devices known as "channel vocoders" which are used for human-to-human communications and which often need to operate in noisy conditions.

2. Description of the Prior Art

Almost all speech recognition systems carry out an acoustic analysis to derive (typically every 10 ms) a "frame" consisting of an estimate of the smoothed short-term power spectrum of the input signal. Such frames are almost always computed using either linear prediction or a bank of band-pass filters. The noise reduction technique described in this invention applies primarily to the latter kind of analysis.

One method of reducing the background noise added to a speech signal in a noisy environment is to use a noise-cancelling microphone. Such an approach, while a useful contribution, is often not enough in itself. It is complementary to the techniques described in this invention, and can be used freely in combination with them.

The remaining methods involve processing the signal, usually in digitized form. These methods can be classified by two criteria: whether they use a single or multiple microphones, and whether they operate on the acoustic waveform or on the short-term power spectrum. This classification results in four possible combinations, and all four have been tried.

Single-microphone waveform-based methods have been tried. They are effective at removing steady or slowly-changing tones, but they are much less effective at removing rapidly changing tones or atonal interference such as helicopter rotor noise.

Single-microphone spectrum-based methods have also been tried. They assume that the noise spectrum is stationary over periods when speech may be present. In one method, the noise spectrum is estimated over a period when there is no speech and then subtracted from the speech spectrum. In another method, the noise spectrum is used to identify frequency bands which will be ignored because they contain a noise level higher than the speech level in the incoming speech or in the particular frame of reference speech against which the incoming speech is being compared.

Multiple-microphone waveform-based methods have also been tried, and with two variations. In the first method, the microphones are used as a phased array to give enhanced response in the direction of the speaker. This, like the use of a noise-cancelling microphone, is an approach that can be combined with the invention described here.

In the second multiple-microphone waveform-based method, which is closely related to the present invention, one microphone (the "speech microphone") collects the speech plus the noise and the other (the "reference microphone") aims to collect only the noise. The noise waveform at the two microphones will, in general, be different, but it is assumed that an appropriate filter (one example being a finite-impulse-response ("FIR") filter) can be used to predict the noise waveform at the speech microphone from the noise waveform at the reference microphone. That is, s_i, the i'th sample of the noise waveform at the speech microphone is approximated by:

$$\hat{s}_i = \sum_{j=0}^{L-1} w_j \cdot r_{i-j}$$

where r_i is the i'th sample of the noise waveform at the reference microphone and w_j is the j'th coefficient of the FIR filter of length L. Adaptive two-channel filtering methods can then be used to design the FIR filter, provided that its characteristics are changing only slowly. The method requires adaptively determining the values of the coefficients in the FIR filter that will minimize the mean-square error between the actual and predicted values of the noise waveform at the speech microphone; that is, the method requires minimizing $\langle e_i^2 \rangle$ where

$$e_i = s_i - \hat{s}_i.$$

This second multiple-microphone waveform-based method works well with single sources of noise, such as a single loudspeaker, but has not been found to be effective with multiple, distributed time-varying noise sources of the kind occurring in aircraft and in many other noisy environments. As an example of the problem faced by this method, consider the situation where the waveform sampling rate is 10 kHz so that the separation in time

between adjacent taps in the filter is 0.1 ms. In this time a sound wave in air travels about one-tenth of an inch, so that if the relative distance between the source of the two microphones changes by even that small distance the filter coefficients will be out by one position. If the filter was accurately cancelling a component in the noise at 5 kHz before the source moved, it will quadruple the interfering noise power at that frequency after the source moved one-tenth of an inch.

Two-microphone spectrum-based methods have also been tried, although not widely reported. If the relationship between the power spectrum at the speech microphone and the power spectrum at the reference microphone can be described by a single linear filter whose characteristics change only slowly, then the noise spectrum at the speech microphone can be predicted from the noise spectrum at the reference microphone as

$$S_{ik} = \alpha_k \cdot R_{ik}$$

where S_{ik} and R_{ik} represent the noise power in the i'th frame and the k'th frequency band for the speech and reference signals respectively. That predicted value of the noise power in the speech channel can be exploited as in the single-microphone spectrum-based method. The advantage of the two-microphone method is that the noise intensity and the shape of the noise spectrum can change during the speech. However, the relationship between the two noise spectra would be determined during a period when there is no speech and must remain constant during the speech.

The limitations of the present art can be summarized as follows. Single-microphone methods operating on either the waveform or the spectrum cannot deal effectively with rapidly time-varying noise. Multiple-microphone methods operating on the waveform cannot deal effectively with moving noise sources. Current dual microphone methods operating on the spectrum cannot deal effectively with multiple noise sources whose effect at the two microphones is different.

The present invention discloses a variation of the two-microphone method operating on the spectrum. It differs from previous methods in using an adaptive least-squares method to estimate the noise power in the signal from the speech microphone from a time-sequence of values of noise power in the signal from the reference microphone. Such adaptive least squares methods have previously been applied only to waveforms, not to power spectra.

Previous methods for estimating noise power directly have either assumed it to be constant and taken an average from the speech microphone over a period when speech is absent, or have used single noise values from a reference microphone rather than taking linear combinations of sequences of such values.

Summary of the Invention

By the present invention, there is provided an apparatus for separating speech from background noise comprising:

means to input speech contaminated with background noise to provide a noisy speech signal

means to input primarily the background noise contaminating the speech to provide a reference signal

signal processing means by which an estimate of the noise power contaminating the speech is obtained by an adaptive least-squares adaptation method from a plurality of recent samples of the power in the reference signal, and

signal processing means by which said estimate of the noise power contaminating the speech is subtracted from the total power of said noisy speech signal to obtain an estimate of the power in the speech.

The present invention is directed to enhancing the recognition of speech which has been detected by a microphone (the "speech microphone") in a noisy environment. It involves a second microphone (the "reference microphone") which has been placed in the same noisy environment so that as little as possible of the desired speech is detected by that microphone. An adaptive least-squares method is used to estimate the noise power in the signal from the speech microphone from a time-sequence of recent values of noise power in the signal from the reference microphone.

The determination of the the estimate of the noise power in the signal from the speech microphone when speech is present is based on the relationship of the noise powers at the two microphones when speech is not present at either microphone.

An adaptive algorithm, known as the Widrow-Hoff Least Mean Squares algorithm, is particularly appropriate for determining (during periods when no speech is present) the coefficients to be used in the linear combination of recent values of noise power in the signal from the reference microphone. However, other known and still-undiscovered algorithms may be acceptable for this purpose.

When speech is present, the previously determined estimate of the noise power in the noisy speech signal is subtracted from the noisy speech signal to leave as the output of the system an estimate of the speech power uncontaminated with noise.

Brief Description of the Drawings

Various objects, features and advantages of the present invention will become apparent from a consideration of the following detailed description and from the accompanying drawings.

FIG. 1 illustrates the hardware which is used in this invention.

FIG. 2 illustrates the processing of the signal in each of the two channels in the DSP chip 7.

FIG. 3 illustrates further processing applied to the reference signal in the DSP chip 7, by which recent values of the power in the reference signal are linearly combined and subtracted from the noisy speech signal to obtain the output of the apparatus.

FIG. 4 illustrates the processes in the DSP chip 7 for determining the coefficients for the linear combination of recent values of the power in the reference signal.

Detailed Description of Preferred Embodiments

Referring to FIG. 1, the invention comprises two microphones 1, 2, a push-to-talk switch 3, two low-pass filters 4, 5, a two-channel analog-to-digital ("A/D") converter 6, and a digital signal processing ("DSP") chip 7. One of the microphones 1 is intended to pick up the speech which is contaminated with noise, and the other microphone 2 is intended to pick up only the noise. The path of the signal and the processing operations related to the signal from the speech microphone 1 will be called the "speech channel", and the path of the signal and the processing operations related to the signal from the reference microphone 2 will be called the "reference channel".

Although the noise at the two microphones is assumed to come from the same set of sources, its form will be different because, for example, the relative intensities of the various sources contributing to the noise will be different at the different locations of the two microphones.

In the speech channel, the signal out of the speech microphone 1 is first directed through a low-pass filter 4, and in the reference channel the signal out of the reference microphone 2 is first directed through a low-pass filter 5. The low-pass filters 4, 5 would be essentially identical. To prevent aliasing upon subsequent digitization, the low-pass filters 4, 5 would have a cut-off frequency of approximately 3.7 kHz.

The signals out of each low-pass filter 4, 5 are next subjected to A/D conversion. Conventionally and conveniently, the system would be provided with a single two-channel A/D converter 6 so that only one such device is required in the system, but alternatively there could be two distinct devices for A/D conversion. The A/D converter 6 would typically sample the two channels at a rate of 8 kHz. It would do this by having a 16 kHz sampling rate and taking samples alternately from the two inputs. The samples should be measured with a precision of 12 bits or better.

The two channels of output from the A/D converter 6, representing the digitized signals from the two microphones 1, 2, are then directed to the two inputs of the DSP chip 7. A suitable DSP chip is model AT&T DSP32C manufactured by American Telephone and Telegraph Company. That chip can be programmed in the high-level language called "C".

The push-to-talk switch 3 is connected to the DSP chip 7. In the case of the recommended DSP chip, this switch would simply be connected to ground when pressed to indicate that speech is present, but the nature of the signal given when the switch is pressed will depend on the requirements of the DSP chip used. The purpose of the switch 3 is to indicate that speech is present at the speech microphone 1 and that therefore the DSP chip 7 should suspend the calculating of the relationship between the noise at the speech microphone 1 and the noise at the reference microphone 2.

In an alternative embodiment of the invention, the switch 3 may be an automatic device which detects the presence of speech at the speech microphone, according to methods well known in the art.

The purpose of the switch 3 is simply to suspend the calculation of the relationship of the noise power at the two microphones when speech is present. Switch 3 is not necessarily used to indicate that the speech recognition system should receive that speech. If the user desires to utter speech that is not intended to be directed to the speech recognition system (called here "extraneous speech"), he must nevertheless press the switch 3 to suspend the calculations just mentioned. An automatic device which detects all speech, extraneous or not, is well suited to that function.

If the speech recognition system should not receive extraneous speech, it will be necessary to have an additional switch to indicate which speech is to be forwarded to the speech recognition system. Therefore, an alternative embodiment of the invention comprises two switches so that one switch (which could appropriately be an automatic device) is used to suspend the calculations of the noise power relationships and another switch is used to send the digitized speech to the speech recognition system which follows after the present invention.

If there is only a simple press-to-talk switch 3 (whether automatic or not) as illustrated in FIG. 1, so that all output of the invention is directed to the speech recognition system, and the user desires to

utter extraneous speech, he should wait a short time (at least a few seconds, but the longer the better) after the extraneous speech before uttering speech that is intended to be reocgnized by the speech recognition system.

The output of the DSP chip 7 will be a digitized representation of the power spectrum of the speech with the noise essentially removed, typically represented by 20 numbers every 8 ms. This output could then be passed to a speech recognition system of a type, well known in the art, which operates on the power spectrum of the speech to be recognized.

FIG. 2 illustrates the processes in the DSP chip 7 with respect to only one of the channels. Identical processes are carried out for both channels. If the channels have been combined by multiplexing at the output of the A/D converter, as is common and appropriate for the preferred DSP chip identified above, the first operation in the DSP chip 7 will be de-multiplexing of the signals.

The incoming signal is written to a first ring buffer containing 256 elements. Every 8 ms, during which 64 samples will have accumulated, the contents of the first ring buffer are copied to another 256-element ring buffer and there multiplied by a Hanning (raised-cosine) window function stored in a 256-element table. Thus, if the n'th element of the first ring buffer is q(n), and the n'th element in the table containing the raised-cosine window function is h(n), the corresponding element in the buffer containing the windowed signal is t(n) where

$$t(n) = q(n) \cdot h(n)$$

A fast Fourier transform is then applied to the 256 values in the second ring buffer. writing the i'th real and imaginary elements of the resulting 128-element complex spectrum as $x_k(i)$ and $y_k(i)$ respectively, where k denotes the k'th block of 64 samples to be transferred, the power spectrum can be computed as $p_k(i)$ where

$$p_k(i) = x_k(i) \cdot x_k(i) + y_k(i) \cdot y_k(i)$$

The 128-element power spectrum must then be grouped into a set of, say, 20 frequency bands. The subscript j will be used to identify these 20 bands. Typically, these bands would be spaced to reflect the frequency resolution of the human ear, such as by having the centre frequencies equally spaced up to 1 kHz and then logarithmically spaced up to the highest band. The power in the j'th band for the k'th block of 64 samples would be computed as

$$b_j(k) = \sum_{i=0}^{127} w_j(i) \cdot p(i)$$

where $w_j(i)$ is the value of a window function forming the j'th band and corresponding to the i'th element of the power spectrum. The values of $w_j(i)$ will be stored in a table in the DSP chip 7. Typically, the window function $w_j(i)$ has the form of a triangle with its apex at the centre frequency of the j'th frequency band and its base spanning the range from the centre of frequency band j-1 to the centre of frequency band j+1, so that the value of $w_j(i)$ is zero outside the range of frequencies covered by the base of that triangle.

The identical processes illustrated in FIG. 2 are carried out for both the speech and reference channels. The power value $b_j(k)$ mentioned above can be considered to be the power in the speech channel; another value, which might be denoted $a_j(k)$ will be calculated to represent the power in the reference channel. However, to now simplify the notation, the subscript j (which indicates that the value pertains to the j'th frequency band) will be dropped because the following operations are carried out for all the frequency bands (typically, 20 bands). Therefore, the power in the reference channel is denoted a(k) and the power in the speech channel is denoted b(k) for the k'th block of samples.

The power in the speech channel, b(k), consists of both speech power and noise power, which can be considered to be additive and which will be denoted by the symbols s(k) and c(k) respectively. That is,

$$b(k) = c(k) + s(k)$$

Referring now to FIG. 3, the values of the noise power in the reference channel are retained in a ring buffer capable of holding the latest M values of a(k). A typical value for M, the number of elements in the ring buffer, is 20. The values of the noise power in this ring buffer are combined linearly to produce an estimate, $\hat{c}(k)$, of the noise power in the speech channel. In other words, the latest M values of noise in the reference channel are reasonably able to predict the current noise in the speech channel. This can be expressed as

$$\hat{c}(k) = \sum_{m=0}^{M-1} \alpha_m \cdot a(k-m)$$

The estimate $\hat{c}(k)$ can then be subtracted from b(k) to form an estimate of the noise-free speech

power $\hat{s}(k)$. That is:

$$\hat{s}(k) = b(k) - \hat{c}(k)$$

Referring now to FIG. 4, the coefficients α_m are derived during periods when no speech is present (as indicated by the switch 3). An adaptive algorithm, known as the Widrow-Hoff Least Mean Squares algoritnm, is used to update the coefficients α_m every time a new value of k occurs (typically, every 8 ms). This algorithm is the following:

$$\alpha_m' = \alpha_m + 2\mu[b(k) - \hat{c}(k)]^{\bullet} a(k-m)$$

where α_m is the m'th coefficient before updating and α_m' is the corresponding value after updating. The initial values of the coefficients α_m can be set to zero.

The constant μ controls the rate of adaptation, a large value giving faster adaptation but less-accurate final values of the coefficients in the case when conditions are stable. The choice of a value of μ therefore should depend on how quickly the noises are changing. Different bands, of which there are typically 20 denoted by the subscript j, can have different values of μ, and in general the values of μ should be related to the standard deviation of the energy values in the speech channel over time.

It is possible for inappropriate values of the coefficient α_m to lead to the illogical result $\hat{c}(k) < 0$. In that event, $\hat{c}(k)$ should be set equal to zero. It is also possible that some calculations lead to $\hat{c}(k) > b(k)$. In that event, $\hat{c}(k)$ should be set equal to b(k).

The output of the apparatus is a set of values of $\hat{s}(k)$ for all frequency bands (typically 20 bands). Previously in this specification, the bands were represented by the subscript j, so the output might appropriately be represented as $\hat{s}_j(k)$. This constitutes an estimate of the noise-free speech power and is well suited to be the input to a speech recognition system that accepts power values as inputs.

The output $\hat{s}_j(k)$ could also be used as the input to a channel vocoder, which is a device for transmitting speech in a digitized form.

The benefit provided by this invention of extracting the background noise will be useful in many types of device intended to either transmit or recognize speech.

Thus, the present invention is well adapted to carry out the objects and attain the ends and advantages mentioned, as well as those inherent therein. While presently preferred embodiments of this invention have been described for purposes of this disclosure, numerous changes in the arrangement of parts, configuration of the internal software, and choice of algorithms will suggest themselves to those skilled in the art. Those changes are encompassed within the spirit of this invention and the scope of the appended claims.

Claims

1. An apparatus for separating speech from background noise comprising: means to input speech contaminated with background noise to provide a noisy speech signal means to input primarily the background noise contaminating the speech to provide a reference signal signal processing means by which an estimate of the noise power contaminating the speech is obtained by an adaptive least-squares adaptation method from a plurality of recent samples of the power in the reference signal, and signal processing means by which said estimate of the noise power contaminating the speech is subtracted from the total power of said noisy speech signal to obtain an estimate of the power in the speech.

2. An apparatus as claimed in claim 1 of which the output of the apparatus, in the form of the estimate of the power in the speech, is connected to a speech recognition system.

3. An apparatus as claimed in claim 1 or 2 in which said adaptive least squares adaptation method uses the Widrow-Hoff Least Mean Squares algorithm.

4. An apparatus as claimed in any preceding claim in which said adaptive least-squares adaptation method combines said samples linearly using coefficients in the combining formula that were previously derived during recent periods when no speech was present in said noisy speech signal.

5. A method of separating background noise from a noisy speech signal comprising continually monitoring background noise to provide a reference signal; processing the reference signal to obtain an estimate of the power thereof using an adaptive least-squares adaptation method from a plurality of recent samples of the power of the reference signal; and processing the noisy speech signal by subtracting the estimate from the total power of the noisy signal to obtain an estimate of the power in the speech.

6. An apparatus which is substantially as herein described in relation to the accompanying drawings.

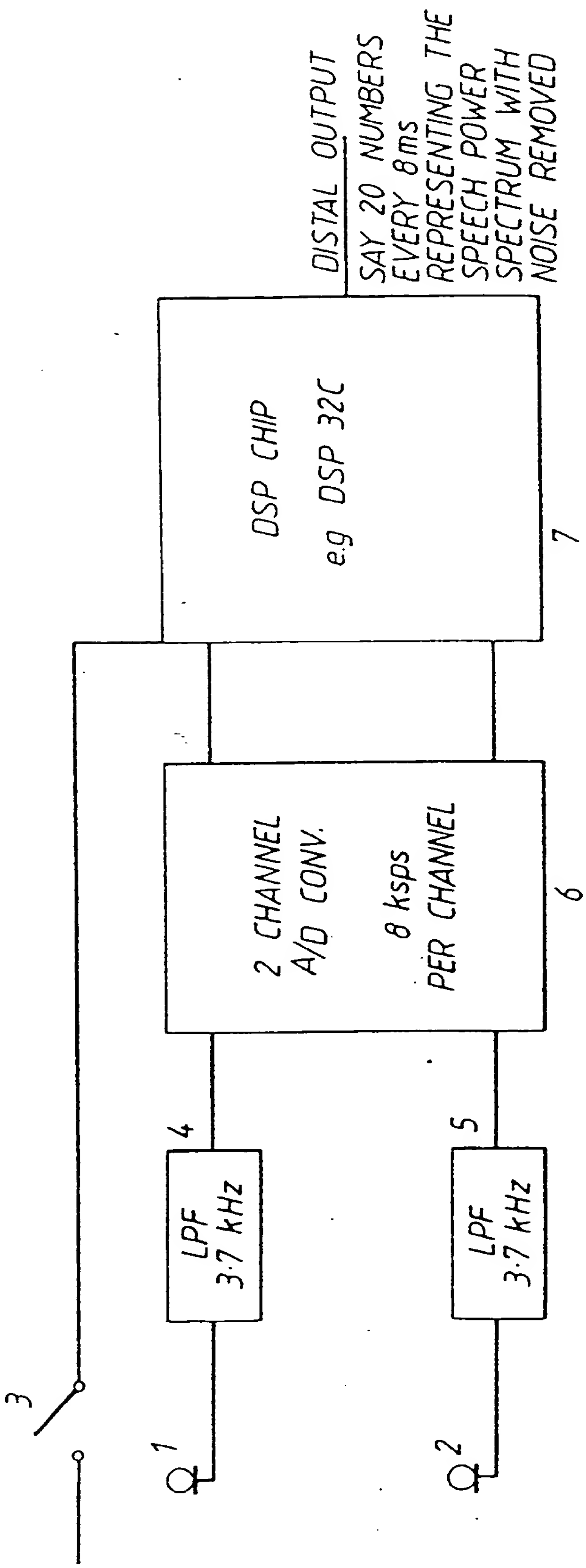
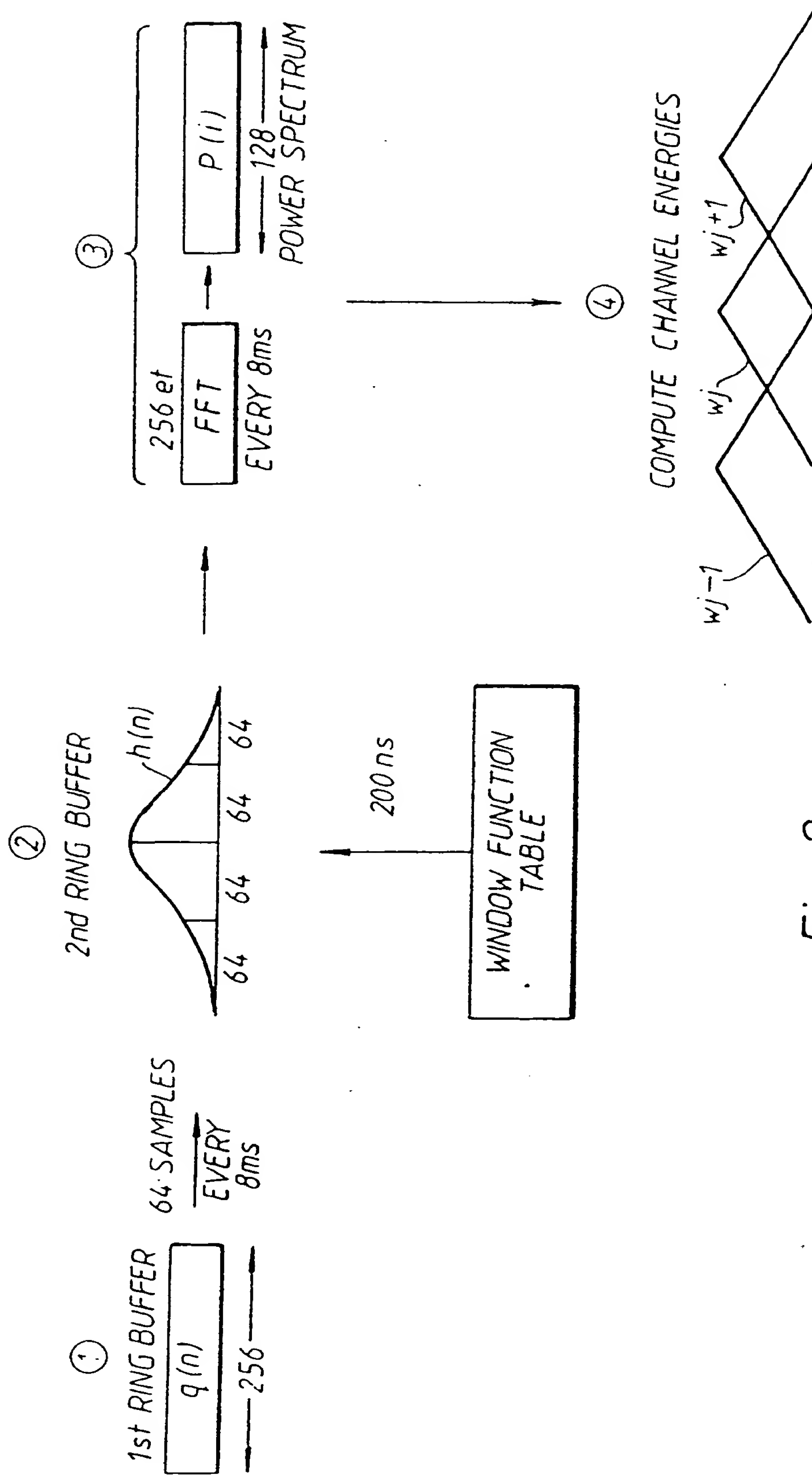

DISTAL OUTPUT
SAY 20 NUMBERS EVERY 8ms REPRESENTING THE SPEECH POWER SPECTRUM WITH NOISE REMOVED
DSP CHIP
e.g DSP 32C
7
2 CHANNEL A/D CONV.
8 ksps PER CHANNEL
6
LPF 3·7 kHz
4
LPF 3·7 kHz
5
3
1
2

Fig.1



1
1st RING BUFFER
q(n)
256
64 SAMPLES
EVERY 8ms
2
2nd RING BUFFER
h(n)
64
64
64
64
200 ns
WINDOW FUNCTION TABLE
3
256 et
FFT
EVERY 8ms
P(i)
128
POWER SPECTRUM
4
COMPUTE CHANNEL ENERGIES
wj-1
wj
wj+1

Fig. 2.

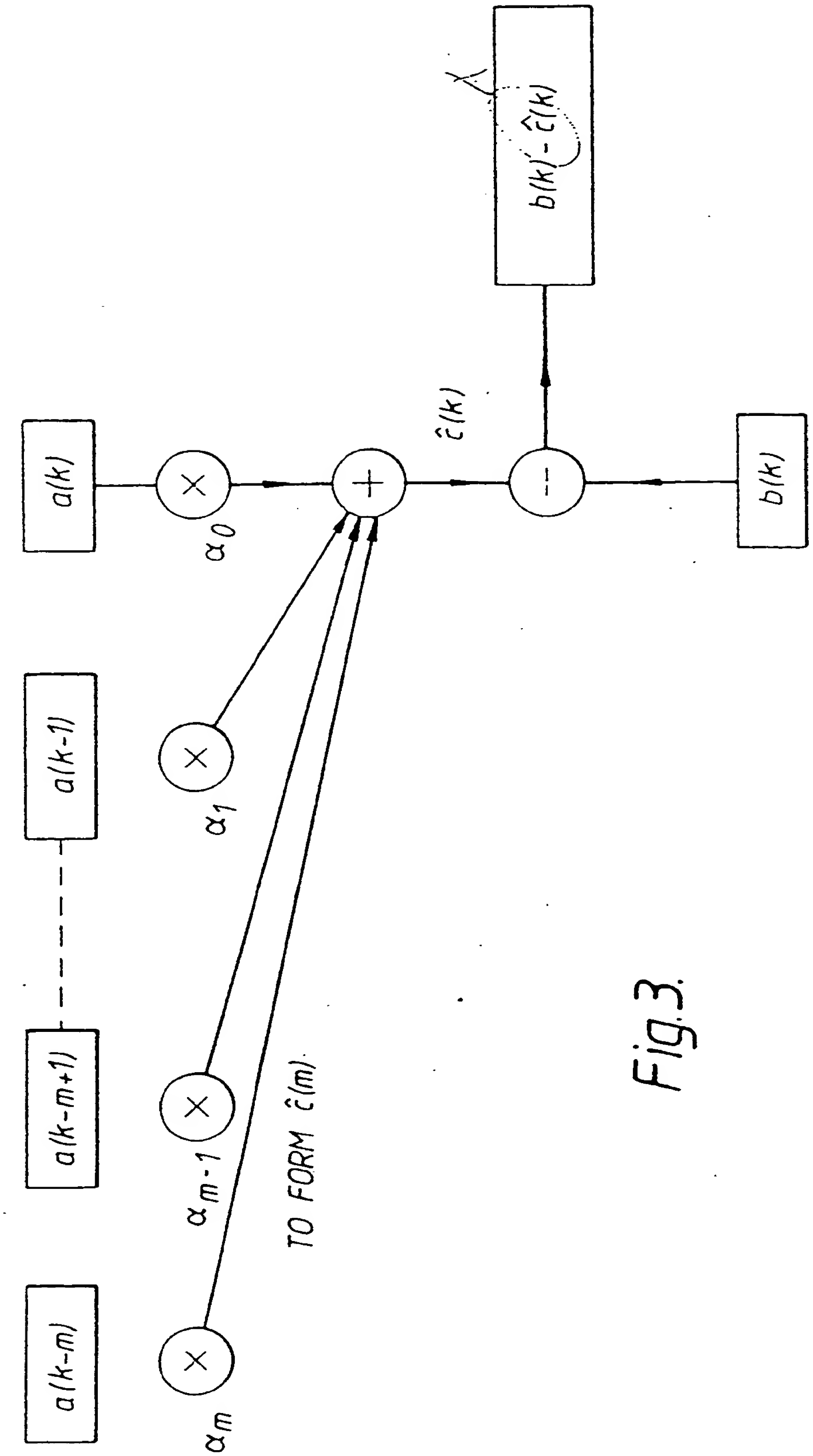


Fig.3.

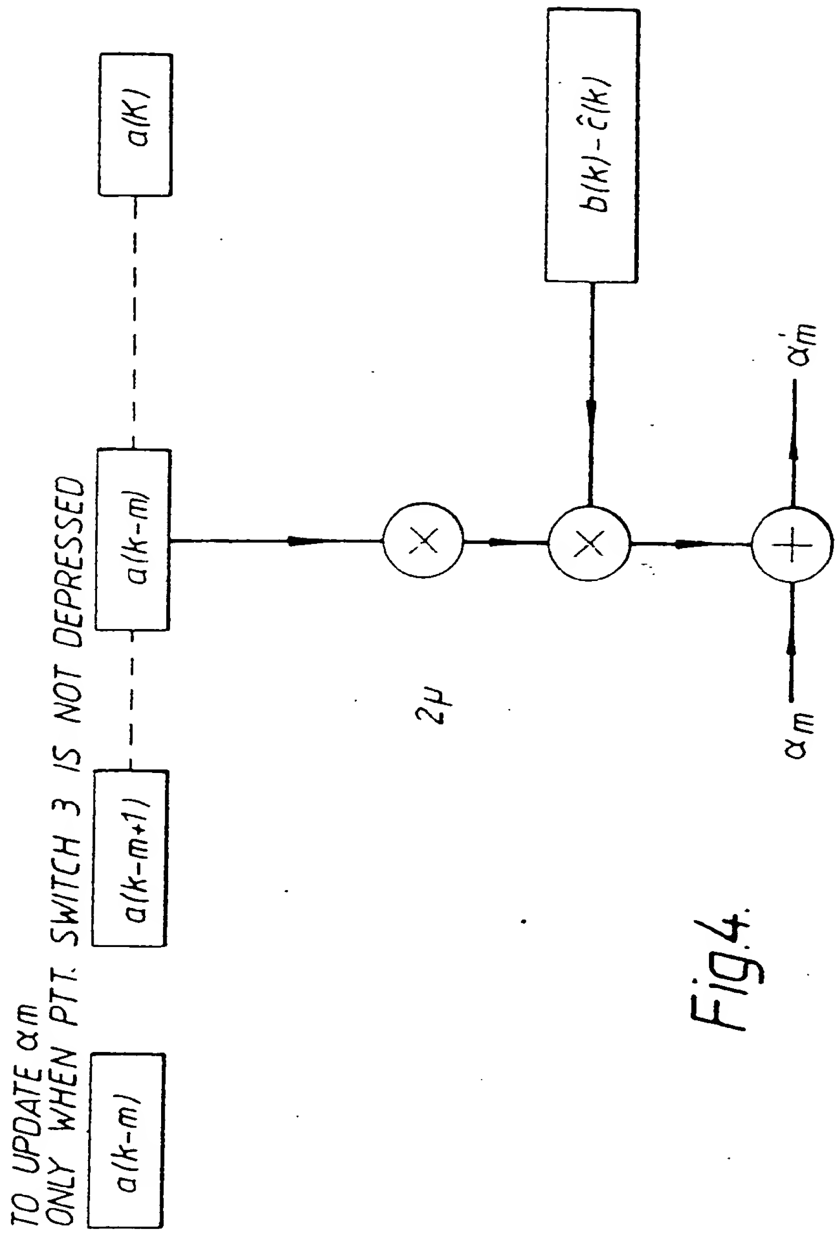

TO UPDATE αm
ONLY WHEN PTT. SWITCH 3 IS NOT DEPRESSED
a(k-m)
a(k-m+1)
a(k-m)
a(K)
b(k)-ĉ(k)
2μ
αm
α'm

Fig.4.

European Patent Office

EUROPEAN SEARCH REPORT

Application Number

EP 91 30 6317

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int. Cl.5)
E	GB-A-2 239 971 (NATIONAL RESEARCH COUNCIL OF CANADA) * the whole document * & CA-A-2 031 641 (NRC OF CANADA) 7 June 1991 ---	1-6	G10L3/00 G10L3/02
Y	US-A-4 932 063 (NAKAMURA) * column 3, line 10 - line 26; figure 1 * * column 3, line 37 - line 44 * * column 4, line 1 - line 32 * * column 5, line 47 - line 58; claims 1-5,8 * ---	1-5	
Y	EP-A-0 332 890 (IBM) * column 5, line 28 - column 6, line 47; figure 2 * * column 7, line 11 - line 40; claims 1-7 * ---	1,2,4	
Y	PROCEEDINGS OF EUSIPCO 88, EUROPEAN SIGNAL PROCESSING CONFERENCE vol. 1, 5 September 1988, GRENOBLE, FRANCE pages 419 - 422; AMIN: 'A generalized noise canceller using orthogonal trasformations' * pages 419-420, " 2.Transform domain noise canceller", " 3.Presetting vs natural setting"* ---	3	TECHNICAL FIELDS SEARCHED (Int. Cl.5) G10L
Y	INTERNATIONAL CONFERENCE ON ACOUSTICS SPEECH AND SIGNAL PROCESSING vol. 3, 14 April 1983, BOSTON MASSACHUSETS pages 1133 - 1136; HOY ET AL: 'Noise suppression methods for speech applications' * the whole document * ---	5	
A	DE-A-4 012 349 (RICOH) * the whole document * -----	1,2	

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	20 MARCH 1992	FARASSOPOULOS A.

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document